Micronas.7435

UNITED STATES PATENT APPLICATION

of

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for

COMPENSATION CIRCUIT AND COMPENSATION METHOD TO COMPENSATE NONLINEAR DISTORTIONS OF AN A/D CONVERTER

Compensation Circuit and Compensation Method to Compensate Nonlinear Distortions of an A/D Converter

The invention relates to a compensation circuit to compensate nonlinear distortions of an A/D converter (A/D: analog/digital) having the features listed in the preamble of Claim 1, and to a compensation method to compensate nonlinear distortions of an A/D converter.

A/D converters are a critical component in integrated circuits having mixed, i.e., analog and digital signal processing. Requirements related to the linearity of the A/D converter are quite difficult to achieve given the usual tolerances for analog components. The measures required for this purpose in the area of analog design entail high cost and/or high current consumption by the circuit.

In order to prevent nonlinear distortions in the analog-to-digital conversion process of the A/D converter, compensation circuits are employed to compensate these nonlinear distortions of the A/D converter, these compensation circuits having an analog signal input. Usually, they are a component on the input side of the A/D converter.

The goal of the invention is therefore to propose a compensation circuit to compensate nonlinear distortions of an A/D converter, which circuit provides for a simplified overall analog design together with, preferably, reduced current consumption.

This goal is achieved by a compensation circuit to compensate nonlinear distortions of an A/D converter having the features of Claim 1 and by a method to compensate nonlinear distortions of an A/D converter having the features of Claim 18.

A compensation circuit to compensate nonlinear distortions of an A/D converter may advantageously have a signal input and a compensation circuit composed of digital circuit elements to digitally compensate nonlinear distortions, the signal input as the compensation input being a digital signal input to supply a signal outputted in distorted form by the A/D converter. Implementation is thus in the form of an output-side digital compensation or distortion correction of the nonlinear distortions from an input-side A/D converter. The digital circuit elements required for this purpose are inexpensive and readily available based on a simple digital circuit design. It is no longer necessary to incur high costs or have increased current consumption based on a compensation circuit added on the input-side of, or integrated into, the A/D converter.

According to the method, after analog-to-digital conversion within the A/D converter and a nonlinearly distorted digital signal is outputted, compensation is implemented in the digital circuit segment.

Advantageous embodiments are the subject of the dependent claims.

An embodiment and modifications thereof will be explained in greater detail with reference to the drawing.

Figure 1 shows basic elements of the compensation circuit to compensate nonlinear distortions of an A/D converter.

Figure 2 shows a circuit to determine a test signal;

Figure 3 shows a circuit for the iterative calculation of correction coefficients; and

Figure 4 shows an alternative circuit for the iterative calculation of correction coefficients using a look-up table.

As is evident from Figure 1, the compensation circuit in the example is composed of an analog section, shown at left, and a digital section shown on the right. An analog input signal x(t) is supplied to an A/D converter ADC/1 (A/D:analog-to-digital). After analog-to-digital conversion, a digital sequence of characters x_n is outputted from the output of A/D converter 1, which sequence corresponds to the analog input signal x(t). When a conventional A/D converter 1 is used, the digital sequence x_n from this converter exhibits nonlinear distortion. The index n represents the sequence of sampling values x_n, n = 0, 1, 2, ...

The A/D-converted sequence x_n is then fed to the input of a compensation circuit 2. Compensation circuit 2 compensates

or corrects the nonlinear distortion which has been created by A/D converter 1. Finally, a sequence of compensated digital data y_n is outputted from compensation circuit 2 as the output data sequence.

In order to be able to implement the compensation, coefficients c_1, c_2, ... c_K, c_0 are supplied to compensation circuit 2 which have been determined or calculated based on the nonlinear distortion response of A/D converter 1. The index k = 1, 2, ..., K here functions as the consecutive index for the coefficients c_k of compensation.

The circuit of this design consisting of A/D converter 1 together with the following compensation circuit 2 to which a set of coefficients c_1, ... c_K is supplied thus offers a simple design which enables an analog-to-digital conversion of an analog signal x(t) to form a sequence of compensated digital data y_n which does not suffer from nonlinear distortion.

Additional components of the circuit, shown in the segment enclosed by the broken line, serve the purpose of determining the coefficients c_1, ... c_K. These additional components are advantageously active only during a configuration phase. As an alternative to the circuit described below, it is also possible to employ a memory in which a set of coefficients c_1, ... c_K, determined previously only once, is stored which may be applied generally for the compensation. The preferred approach, however, is the circuit design described below which provides an adjustment of the coefficients c_1, ... c_K to the actual and/or instantaneous conditions.

In the circuit shown in the example, a test signal s(t) is generated by a test signal generator 3 and applied during the configuration phase to the input of A/D converter 1. Test signal generator 3 also provides either the parameters s_n to generate the analog test signal s(t) or a sequence of digital test signal data S_n corresponding to this analog test signal s(t). These parameters s_n or test signal data S_n are fed to a coefficient determination system 5 and/or to a test signal check device 4.

In addition, the circuit has a coefficient determination system 5 to determine the coefficients $c_1, \ldots c_K$ to be used by the compensation circuit. In addition to an output to transmit the determined coefficients $c_1, \ldots c_K, c_0$ to compensation circuit 2, coefficient determination system 5 has an input to which the sequence of digital data x_n from A/D converter 1 is supplied. In addition, coefficient determination system 5 has an input to which distortion data or difference data D_n are supplied. The difference data D_n are provided by a subtracter 6 to which the sequence of compensated digital data y_n is fed from the output of compensation circuit 2. Subtracter 6 also receives a sequence of digital signal data S_n either directly or indirectly through test signal check device 4. The sequence of digital signal data S_n corresponds in the configuration phase to a signal map of the test signal, whereby, after appropriate determination of coefficients has been effected, this signal map is as undistorted as possible, or ideally is completely undistorted.

During the configuration phase, the analog test signal s(t) is thus generated and fed to A/D converter 1. The converter implements an analog-to-digital conversion for

the sequence of digital data x_n that is supplied both to compensation circuit 2 and coefficient determination circuit 5. If the set of coefficients $c_1, \ldots c_K$ is not available, compensation circuit 2 outputs the sequence of digital data $x_n = y_n$ as the sequence of compensated digital data y_n. This data is fed to subtracter 6 that is also supplied with a corresponding sequence of digital test signal data S_n which matches an undistorted data set. After subtraction of the two sequences of data ($y_n - S_n$), the data sequence of the difference signal D_n is supplied to coefficient determination system 5. Using this data, coefficient determination system 5 determines the set of coefficients $c_1, \ldots c_K, c_0$ that is supplied to compensation circuit 2 for future compensation.

According to the preferred embodiment, a test signal s(t) is subsequently sent by test signal generator 3, which may also be composed of a memory with an analog test signal s(t) and a corresponding digital parameter set s_n, to A/D converter 1. The sequence of digital data x_n generated by A/D converter 1 is subsequently compensated by compensation circuit 2 in accordance with this supplied set of coefficients $c_1, \ldots c_K$, such that the sequence of compensated digital data y_n is outputted, ideally with already optimized coefficients $c_1, \ldots c_K$, without nonlinear distortion. The sequence of compensated digital data y_n is in turn fed to subtracter 6 in which, after subtraction using the corresponding values of the sequence of digital test signal data S_n, the sequence of data D_n of the difference signal is again generated. This difference data D_n is again fed to coefficient determination system 5 which, in the event difference

data D_n does not equal zero or exceeds predetermined threshold values, implements another, or preferably, an iteratively improved determination of coefficients so as to provide improved coefficients $c_1, \ldots c_K$.

After a sufficiently distortion-corrected or compensated set of coefficients $c_1, \ldots c_K$ has been determined, the configuration phase ends, after which the circuit composed of A/D converter 1 and compensation circuit 2 implements a conversion and compensation of analog signal x(t) to form a sequence of compensated digital data y_n.

Appropriately, a configuration phase is initiated at regular intervals in order to check the set of coefficients used, $c_1, \ldots c_K$ in terms of their current validity. In this way, drifting nonlinear distortions caused, for example, by the circuit heating up or other interfering effects from the environment can be compensated.

The configuration phase is turned on and off a sufficient number of times as needed so that any slow changes in nonlinear distortions can be detected early enough and then compensated. It is, of course, in principle also possible to implement a predetermination and consideration of anticipated additional degradations or improvements related to the generation of distortions.

A control device C is advantageously employed to control the compensation circuit, the control device being connected to a time-monitoring device, specifically, a timer T. In addition to turning the configuration phase on and off, control device C also controls the individual components through, for example, bus 7.

In particular, it is possible to provide different types of test signals s(t), s_n for different application areas of the circuit so that coefficients c_1, ... c_K may, for example, be optimally adjusted for a low-frequency or high-frequency analog signal x(t).

The following discussion examines in more detail the circuit components and the operational sequences of the method with reference to the mathematical background.

The configuration phase starts with an analog test signal

s(t),

to which the theoretical uncorrupted sampling values

$$S_n = s\left(\frac{n}{F_S}\right),$$

correspond after analog-to-digital conversion, without nonlinear distortion, which values are in turn to be outputted after optimal compensation from compensation circuit 2 as the sequence of compensated digital data y_n. Here n is a consecutive index of the set of natural numbers, while F_s is the sampling frequency of A/D converter 1.

After the analog test signal s(t) to A/D has been fed to converter 1 and the analog-to-digital conversion has been performed in the converter, the sequence of

digital data x_n is supplied to its output based according to the expression

$$x_n = s_n + d_n.$$

The sequence of digital data x_n thus corresponds to the summation of correct theoretical and uncorrupted sampling values s_n, and the respective distortion data value d_n which matches the corresponding distortion by A/D converter 1.

Digital compensation circuit 2 which uses the coefficients $c_1, \ldots c_K$ to generate the sequence of compensated digital signals y_n that are ultimately to be outputted must therefore perform a compensation having the characteristic which may be described by a K^{th}-order polynomial:

$$(1) \qquad y_n = \sum_{k=1}^{K} c_k \cdot x_n^k = \nu \cdot s_n + D_n = S_n + D_n .$$

Here the coefficient values c_k with k = 1, 2, ... K are adaptive coefficients, i.e., coefficients which may be adjusted as necessary. The output signal, or sequence of outputted compensated digital data y_n contains a map of the test signal or of the sequence of digital test signal data S_n plus a possible change ν,specifically, amplification or ~~distortion~~ attenuation. The sequence of compensated digital data S_n is thus the product of a distortion factor ν and the sequence of digital test signal data s_n which may be described by

$$S_n = \nu \cdot s_n ,$$

where the effective distortions of the switching sequence of A/D converter 1 and, in the event of insufficient compensation, of compensation circuit 2 may be described by the sequence of difference data D_n according to the expression

(2) $D_n = y_n - S_n$.

Since the parameters s_n of the test signal are known, the sequence of output signal data S_n may be extracted from the sequence of compensated digital data y_n by test signal check device 4 so as to enable the actual distortion data or difference data D_n in data y_n to be calculated at the output of compensation circuit 2. In addition, the gradients of the rms distortion may be calculated using the expression

$$\frac{\partial D_n^2}{\partial c_k} = 2 \cdot (y_n - S_n) \cdot x_n^k = 2 \cdot D_n \cdot x_n^k .$$

The use of an iterative method allows the set of coefficients c_k to converge. To this end, the formulation

(3) $$c_k^{n+1} = c_k^n - G \cdot D_n \cdot x_n^k$$

may be selected so as ultimately to minimize the rms distortion or output. A parameter G is introduced in equation (3) as a stability criterion, which parameter at the same time provides for the highest possible convergence rate. The term c_k^n here describes the value of the coefficients c_k in the n^{th} iteration step, the coefficient of compensation c_k again having the consecutive index k = 1, ..., K. The iteration steps are preferably counted from the value zero, so that n = 0, 1,....

In an especially preferred embodiment, a sinusoidal test signal $s(t) = \sin[2\pi t]$ is preferably used as the test signal to perform the nonlinear compensation since it is then simpler for A/D converter 1 to determine the structure of the distortions, and is possibly simpler for compensation circuit 2 to determine additional distortions. A nonlinearly distorting A/D converter 1 having a sinusoidal input signal at a frequency F_t produces harmonics on frequencies $p \cdot F_t$, which may be folded back by sampling to form

$$f_k = \begin{cases} (p \cdot F_t)_{\bmod F_s} & (p \cdot F_t)_{\bmod F_s} \le \dfrac{F_s}{2} \\ F_s - (p \cdot F_t)_{\bmod F_s} & (p \cdot F_t)_{\bmod F_s} \ge \dfrac{F_s}{2} \end{cases}.$$

Here $p = 2, 3, \ldots, M$ is the consecutive index of the frequency calculation.

If the first harmonics are significant, specifically, if $p = 2, \ldots, M$ applies, the test frequency F_t should be selected such that the frequency band 2B is maximized around the fundamental where none of the first M harmonics fold back according to the expression

$$(4) \qquad B = \max_{F_t} \left\{ \min_{p=2,\ldots,M} \{|F_t - fp|\} \right\}.$$

The extraction of the test signal or sequence of output signal data S_n may be

implemented using a known method of carrier processing. Another factor which must be taken into account is that the test signal s(t) or the sequence of output signal data S_n assigned to this signal are amplitude-conforming.

To this end, a circuit may be used that is based on an I/Q demodulator (I: in-phase, Q: quadrature phase) and a Cordic circuit 43, and calculates the phase and amplitude of the input sequence of digital data y_n. The illustration in Figure 2 provides an example of this circuit.

Frequency, amplitude and direct current (DC) are recovered by feedback using the method known from automatic control engineering. A complete PI control (PI: proportional and integral components) is employed here to determine the frequency. To determine amplitude and direct current, a ~~T~~P-control is used in which only the proportional components are employed. The control parameters used are the P-component of the amplitude control C_a, the P-component for the DC components C_{dc}, and the P-component and I-component of the frequency control C_p or C_i, which components are intended at the same time to meet the stability criterion for a control loop and to ensure the fastest possible transient.

The sequence of digital data y_n outputted by compensation circuit 2 is fed to a test signal check device 4, the supplied sequence of digital data y_n being fed to two multipliers 41a, 41b. A sinusoidal signal $\sin[2\pi t]$ as the signal sequence to be multiplied is fed to the first of the multipliers 41a by a sinusoidal tone generator 41. Analogously, a cosinusoidal signal sequence

cos[2πt] is supplied by the sinusoidal tone generator to second multiplier 41b. After multiplication, the two data sequences are each fed to a filter 42 with undersampling P. After filtering, the I-separated and Q-separated signal components are supplied to Cordic circuit 43 which determines, for the data y_n outputted from compensation circuit 2, a corresponding amplitude and corresponding phase that are supplied through two outputs.

The signal sequence and data sequence representing the amplitude are fed to a series of components 44. These are composed of a subtracter 44a, a multiplier 44b, an adder 44c, and an inverter(z^{-1}) 44e, the output signal of which is fed to subtracter 44a and adder 44c. A coefficient value C_a is supplied as the control parameter for the P-component of amplitude control to the multiplier. The output signal of this series of components is also supplied to another multiplier 44e, to the input of which the cosinusoidal signal cos[2πt] from sinusoidal tone generator 41 is applied. This system is ultimately used to determine the sequence of output signal data S_n that is then supplied to the difference-forming subtracter 6, which in turn has the sequence of digital data y_n applied to it through the second input.

In addition to amplitude, Cordic circuit 43 also outputs a corresponding phase or sequence of phase data. This data is also fed to a circuit composed of a plurality of components 45. In the embodiment shown, this circuit is

composed of two parallel multipliers 45b, 45a to which additionally a coefficient value C_p or a coefficient value C_i is supplied. The output signal of the last-named multiplier 45a is fed to an adder 45c, the output signal of which is supplied both to an inverter (z^{-1}) 45d and to a second adder 45e. The output signal of inverter 45d is fed to the second input of the first-named adder 45c. Through another input, second adder 45e receives data from multiplier 45b in which coefficient value C_p is up-multiplied for the phase data. The output data from this adder 45e are fed to another adder 45f which has two additional inputs. A frequency ratio F_t/F_s consisting of the test frequency F_t of A/D converter 1 and the sampling frequency is supplied through the first additional input. An output value of the inverter (z^{-1}) 45g connected after adder 45f is fed through the second input. Output values of this inverter 45g are supplied as the timing variable t to an input of sinusoidal tone generator 41.

In addition, a DC component DC_n is filtered out and extracted from the sequence of digital compensated data y_n outputted from compensation circuit 2. To this end, the digital compensated data y_n are supplied to a circuit 46 composed of a subtracter 46a, a multiplier 46b, a adder 46c, and an inverter (z^{-1}) 46d. A control parameter C_{dc} is applied through the second input to multiplier 46b. Control parameter C_{dc} determines the rate of the transient. The outputted data sequence from inverter 46d is supplied to

the inputs of subtracter 46a and adder 46c, as well as to the subtracter 6 generating the difference data D_n.

In the circuit of Figure 2, the frequency F_t is thus derived using a phase-locked loop (PLL). The input signal corresponding to the sequence of compensated digital data y_n is split into I-component and Q-component that are then filtered by a low-pass filter 42 of critical frequency B and undersampled. From the filtered I-component and Q-component, Cordic circuit 43 then computes the amplitude and phase between input signal y_n and the locally generated sinusoidal tone from sinusoidal tone generator 41. The phase is passed to the PI control as the error signal. After a settling time, sinusoidal tone generator 41 will generate in its cosine branch a signal synchronized with the test tone. Coefficients C_p and C_i, and test frequency F_t as the known parameters determine the PI control.

The amplitude is derived iteratively from the amplitude output of Cordic circuit 43 using the control parameter C_a. The DC components are filtered and extracted from the output signal using control parameter C_{dc}. The complete circuit composed of test signal check device 4 and following subtracter 6 finally generates a sequence of difference data D_n according to equation (3) proportional to the nonlinear distortions.

Advantageously, available carrier-processing systems and carrier-processing methods may be utilized for the purpose of implementing this circuit. What must be added are the circuits for the amplitude and DC components.

An especially preferred embodiment of compensation circuit 2 is constructed segment-by-segment, as shown in Figure 3.

The data sequence x_n outputted by A/D converter 1 is fed in compensation circuit 2 to a parallel system of multipliers 21. At the same time, the same input signal, i.e., once again the corresponding data value of data sequence x_n, is fed to the second input of first multiplier 21_2 so that a squaring is effected. The output of this first multiplier 21_2 is supplied to the input of the second multiplier 21_3, and so on, such that at each subsequent stage the exponent is increased by the value 1 up to a value x_n^K.

As a result, an exponentiation is effected, where each exponentiation step has an output so that values for digital data with exponentiations x_n^1, x_n^2, ..., x_n^K are outputted from the input and the field of multipliers 21. These are then fed to another field of multipliers 22, whereby a multiplication is performed with one each of the corresponding coefficients ~~$c_k(m)$~~$c_k(m)$ where k = 1, 2, ..., K. What is described is thus a multi-element system with the coefficients $c_k(m)$of the compensation in the m^{th} segment with m = 0, 1, 2, ..., N-1 as the m^{th} segment of the amplitude range. Accordingly, the nonexponentiated value or nonexponentiated data sequence x_n^1 as

well as the coefficient $c_1(m)$ are entered in the first multiplier 22_1 of the second multiplication field 22. The once exponentiated data value x_n^2 and the second coefficient $c_2(m)$ are entered in the second multiplier 22_2, etc. The output values of multipliers 22_1, 22_2, ..., 22_K of second multiplication field 22 are fed to an adder 23 which performs an addition of all input values, and also of the zeroth coefficient $c_0(m)$, then outputs the sequence of compensated digital data y_n.

The sequence of digital data x_n outputted by A/D converter 1 is also fed to a coefficient determination system 5, where a rounding operation is performed in an index determination device 51, taking into account the N segments during the determination of the index m. Here the sequence of digital data x_n with its respective value increased by 1 is divided by 2, then multiplied by the number of segments N. The thus generated segment index m is fed to a coefficient memory system 52 that is composed of a plurality of m parallel memory components 52_0, 52_1, ..., 52_{N-1}. The respective coefficients $c_1(m)$, $c_2(m)$, ..., $c_K(m)$ und $c_0(m)$ are stored in the individual segments of this coefficient memory system 52. For each stored index m, there is an output to an adder 53_1, 53_2, ..., 53_K, 53_0, the output of which in turn is fed back to the same segment of coefficient memory system 52. The result from a multiplier 54_1, 54_2, ..., 54_K is entered in the second input of adders 53_1, ... Each correspondingly exponentiated value of the sequence of digital data x_n^1, x_n^2, ..., x_n^K is fed to the inputs of these multipliers 54_1, 54_2, ..., 54_K. Supplied to each second input of multipliers 54_1, 54_2, ..., 54_K is the result of a multiplier 55 to which both the corresponding

values for difference data D_n and the negative parameter -G serving as the stability criterion are supplied. Only the value for the difference data D_n, multiplied by the negative parameter -G, is supplied to adder 53_0 at the adder's second input.

Using a circuit of this type, compensation circuit 2 may be implemented on a segment-by-segment basis. The range of input data values, that is , the sequence of digital data x_n, from A/D converter 1 with data values from -1 to +1 is, uniformly distributed among N segments according to the expression

$$-1+\frac{2}{N}m \le x_n \le -1+\frac{2}{N}(m+1),$$

where the segment index m lies between 0 and N-1. As a result, one coefficient set $\{c_0(m), c_1(m), ..., c_K(m)\}$ is assigned to each segment. Based on segmental interpolation of the characteristic, zeroth coefficients $c_0(m)$ are added in, so that for the sequence of compensated digital data y_n outputted from compensation circuit 2 the following expression applies:

$$(5) \qquad y_n = \sum_{k=0}^{k} c_k(m)\cdot x_n^k \text{ where } m = \left\lfloor N\cdot\frac{x_n+1}{2}\right\rfloor,$$

where $\lfloor\ \rfloor$ represents the rounding operation.

The equation (5) for compensation in compensation circuit 2, and the iterative calculation of coefficients according to equation (3) may be effectively implemented together, as Figure 3 illustrates. In a memory of size N x (K+1), here coefficient memory system 52, N sets of K+1 coefficients each $c_1(m)$, $c_2(m)$, ..., $c_K(m)$, and $c_0(m)$ are

stored for the respective m = 0, 1, 2, ..., N-1. For each sampling instant, the index m is derived according to equation (5) from the input signal, i.e., from the applied value for the sequence of digital data x_n, and assigned to the corresponding coefficient set 52_0, 52_1, ..., 52_{N-1}, then applied accordingly within compensation circuit 2. Also in this procedure, the stored value for each coefficient is iteratively improved according to equation (3) and stored in the same memory location.

As shown in Figure 4, in a certain case the method with segmental interpolation may be simplified with $N = 2^B$. For this purpose, it is necessary that the number N of segments agree with the resolution of the signal or the sequence of digital data x_n, i.e., $N = 2^B$ applies, where B is the number of bits per sampling value given a signal range between -2^{B-1} and $2^{B-1}-1$ and only one coefficient per segment, i.e., K = 0.

The circuit of Figure 3 is thereby reduced to the circuit of Figure 4 which has, following index determination device 51 for determining the index m, a look-up table with 2^B adaptive coefficients $c_0(m)$in memory fields 52_0, ..., 52_{N-1}. In this arrangement, all multipliers 54 are eliminated so that only the first addition stage with adders 53_1, ... remains, as described above.

Initial synthetic calculations using a sinusoidal tone as the test signal s(t)and a mathematical model of the analog-to-digital characteristic with frequencies $F_s = 40.5$ MHz, $F_t = 1.84$ MHz, the A/D model with coefficients for the calculation based on $0.9895x + 0.0028x^2 + 0.024x^3 - 0.0064x^4$ and for compensation N = 1 segments, and a

maximum running index of m with K = 4, produced significant improvements. After analog-to-digital conversion, the initial value was -60.3 dB, and after compensation the value was 80.9 dB – yielding a significant improvement of 20.5 dB. In the case of one measurement (K3), the improvement found was -47.2 dB after A/D conversion, as compared with -56.3 dB after compensation, giving an improvement of 9.1 dB. The ratio of distortions to outputs after A/D conversion was -46.8 dB, as compared with -55.3 dB after compensation – producing an improvement of 8.4 dB.

In place of a plurality of individual components, as described above, an implementation is also possible in an analogously capable computer chip, or in a monolithically fabricated semiconductor device in the form of an integrated circuit or the like.